

TERRY BEARD
CHAIRMAN
CHIEF EXECUTIVE OFFICER
DIGITAL THEATER SYSTEMS I.P.

Federal Communications Commission
1919 M Street, N.W.
Washington, D.C. 20554

DOCKET FILE COPY ORIGINAL

RECEIVED

AUG 6 1996

5

250

August 5, 1996

Dear Sirs:

Pursuant to applicable procedures set forth in Section 1.415 and 1.419 of the Commission's Rules, 47C.F.R. Sections 1.415 and 1.419, Digital Theater Systems wishes to formally file comments in addition to the comments we filed on July 9, 1996.

Enclosed please find one original and eleven copies to be distributed to each of the Commissioners.

Sincerely,

Terry Beard /
Chairman, C.E.O.

Enclosures
TB/mac

0411

ММВ

TERRY BEARD
CHAIRMAN
CHIEF EXECUTIVE OFFICER
DIGITAL THEATER SYSTEMS, L.P.

DOCKET FILE COPY ORIGINAL

FILE
FCC
AUG 6 1996
L-102

The DTV Open Platform Audio Standard

This is a brief discussion of a proposed Digital Television open platform audio standard before the FCC. This proposed standard takes advantage of the essence of digital technology: programmability. The standard allows for the transmission of the algorithms that decode the digital audio along with the audio data. The amount of algorithm data needed is infinitesimal compared to the amount of audio data and it can be easily stored and updated as needed on nonvolatile memory in the receiver. The standard specifies a minimum platform capability not a specific use of that capability. This is a flexible standard that provides a structure for current manufacture with the essential feature of future backward compatibility, yet takes into account the rapid ongoing technical innovation of the digital era. However, the single most important feature of this proposed standard is that it makes compatibility with other new media practical. On the other hand, the adoption of the ATSC AC3 standard will require that every TV manufacturer and every broadcaster will forever have to support a fixed already obsolete technology. This self imposed dollar and bandwidth cost will be an **unnecessary** burden on them and the consumer.

The key idea is the codestream principle which allows for continuous or periodic updates of improvements in technology in a manner transparent to the consumer. This principle makes use of general purpose components in the design of TV sets, all of which are common in consumer electronic products and therefore offer significant cost reductions to both the manufacturer and consumer.

For broadcasters, the open platform standard assures them they will be competitive and compatible with alternative entertainment delivery means as audio technology advances.

For manufacturers the open platform standard reduces their costs by allowing them to use **general purpose universal platform** chips produced by many competitive parts makers. It enhances the "feature" capability of their product which will be able to automatically take advantage of audio technology advances without having to be redesigned.

For consumers an open platform standard will provide long term elimination of market confusion because they can be assured their set will not need to be upgraded or replaced to take advantage of new audio technology and the competition provided by the open platform standard will assure them the lowest cost possible.

Thank you for your attention to the open platform audio standard concept.

Terry Beard



**Before the
Federal Communications Commission
Washington, DC 20554**

REC-2
FCC
AUG 6 1996
C-100

In the matter of

Advanced Television Systems)
and Their Impact upon the)
Existing Television Broadcast Service)

MM Docket No. 87-268

DOCKET FILE COPY ORIGINAL

Additional Comments of Digital Theater Systems, LP

Digital Theater Systems(DTS) hereby submits additional comments in response to the Fifth Further Notice of Proposed Rule Making("Fifth Further Notice") adopted on May 9,1996 and released on May 20th, 1996 by the Federal Communications Commission("Commission").

Codestream

DTS is proposing that the ATSC standard

- specify and mandate a method for embedding audio decoding algorithms within the MPEG-2 transport layer
- specify a minimum target hardware audio decoding capability

Introduction

The proposed ATSC standard mandates the use of a single proprietary audio decoding algorithm (Dolby AC-3). The DTS proposal does not support the standardization of a single algorithm. We believe that no single algorithm can cover the range of functionality's and applications proposed for advanced digital television services. Moreover, advances in semiconductor technology in terms of memory and processing is opening up algorithmic opportunities that were impractical just a few years ago.

Against this backdrop we believe it is not necessary to standardize on a single audio decoding algorithm, when cost-effective systems can be built to switch between algorithms, or in the future learn new ones. This capability will allow the ATSC standard to keep abreast of future advances in audio coding techniques

Clearly, establishing any primary, or default, audio coding standard, without establishing and mandating a means of over-riding it in consumer TV receivers, will also cripple competition and innovation in this field.

The main emphasis of the DTS proposal is that a method be established and standardized, herein termed **Codestream**, that allows competing audio decoding algorithms to be equally accessible to consumers.

Codestream

Codestream refers to the transmission of the audio decoding algorithm source code embedded within the MPEG-2 transport layer. This can easily be provided within the current ATSC standard as auxiliary data. The Codestream is transmitted simultaneously with the audio data stream, such that proper decoding of the audio signal is achieved by extracting the algorithm from the Codestream.

To ensure the principle of equal access to the consumer of competing proprietary algorithms, it is necessary to mandate a reasonable minimum target decoder. The target decoder demultiplexes, translates and runs the embedded algorithm code to reconstruct the audio data. With the minimum decoder specification readily available, algorithm providers can easily develop compatible high quality audio coding systems which are guaranteed to run on all ATV receivers.

This is similar to the situation in the computer industry where independent software vendors develop and market products that are guaranteed to run provided they are compatible with the application program interface (API).

The DTS Codestream proposal requires

- a standard (but extendible) set of universal processing instructions
- specification of Codestream as an ancillary data service within the MPEG-2 transport layer to carry the coding instructions
- a standard minimum target decoder

Universal Processing Instructions

All current dsp hardware platforms implement a small number of common instructions. These common instructions can form the basis of a universal assembly language that is easily translated into proprietary assembly language specific to particular processing hardware. This core instruction set can be readily agreed upon by the major US silicon manufacturers. This instruction set will of course include options for future extensions.

Ancillary Data Service

The current ATSC standard carries the AC-3 compressed audio data as an ancillary data service within the MPEG-2 transport layer. The DTS Codestream proposal would expand the use of the user data field in order for it to carry the universal audio coding instructions and compressed audio data. The audio coding instructions would be specified as the audio Codestream and would be periodically transmitted as an ancillary data service. The bandwidth of the audio Codestream would be dynamic, and retain compatibility with the flexible channel allocation of the current standard. This would allow broadcasters to determine the update rate of specific coding algorithms irrespective of the file size of the algorithm.

For example current multichannel audio decoding algorithms are approximately 30 Kbytes in size for typical digital signal processors. For these algorithms to be fully updated every 10 seconds the Codestream would need a bandwidth of 24 kbps. This bandwidth could be reduced by, for example, lowering the update rate of the algorithm, layering the Codestream algorithm such that the non-essential routines are updated less frequently, or by taking advantage of redundancy within the decoding program. More importantly since the Codestream is not synchronous to any transmitted audio or video data, it can be transmitted at a variable bit-rate. Given that the compressed video data bandwidth is also variable, it is clear that the Codestream could be efficiently transmitted in bursts during periods of low video activity (low bandwidth) without any increase in overall data bandwidth. Such a scheme could improve the update period dramatically without any effect on the video or audio quality.

The issue of a finite update period is only relevant when the the Codestream algorithm is not already resident in the receiver which will rarely be the case. If the decoding hardware has previously used the algorithm, and has it stored in memory, it will immediately begin to decode the audio data. However if it is the first time the receiver has encountered this particular algorithm, the audio decoder would enter a setup phase during which the new algorithm or upgrade would be downloaded into the hardware. This downloading latency time would be selected by the broadcaster and could range from a fraction of a second to several seconds.

In summary the performance of the Codestream approach is dependent on three factors, the size of the decoding algorithm, the bandwidth allocated to the Codestream, and the amount of memory available for storing algorithms in receivers

Minimum Target Decoder

To ensure that all proprietary audio decoding algorithms can compete fairly within the ATV service, it is necessary to mandate a reasonable minimum target audio decoder.

The target audio decoder demultiplexes, translates and runs the embedded algorithm code to reconstruct the audio data, and is comprised of an instruction/data demultiplexer, an instruction translator and an audio signal processor. The demultiplexer and translator can be internal or external to the processor. The demultiplexer unpacks the Codestream, and passes the universal instructions to the translator. The translator is unique to the processor, and translates the universal instructions into processor specific assembly language. The processor is then able to execute the audio decoding algorithm to produce the sound signals.

A 'reasonable' minimum target decoder should be defined as one which can implement and decode one complete main audio service as defined in the current ATSC standard, using any current commercially available multichannel decoding systems e.g. AC-3, DTS or MPEG.

The key element in the target decoder is the digital signal processor which executes the audio decoding algorithm. Most major US semiconductor companies are already in a position to provide a compatible processor which is capable of running any of these algorithms within the minimum target decoder specification. These companies include Motorola, Texas Instruments, Analog Devices and Crystal Semiconductors.

The minimum target decoder will need to specify certain benchmarks with respect to digital signal processing performance. These include execution speed, internal precision, and an appropriate memory map. For example a complete 5.1 channel main audio service would typically require a 40 MIPS dsp core, with 32 Kbytes of SRAM, 64 Kbytes of ROM, and with an internal architecture capable of 20 to 24-bits single precision

Conclusion

The ATSC DTV audio standard is already technically obsolete in that the mandatory audio coding system (Dolby AC-3) is inferior to more recently developed algorithms. Over time this obsolescence will become more pronounced and could affect consumer adoption of advanced digital television services

By adopting a Codestream approach to the audio standard, in conjunction with a programmable target audio decoder, the ATSC DTV audio standard is protected from obsolescence. The modified standard will also:

- stimulate competition by encouraging innovation in audio coding techniques
- ensure that consumers continue to benefit from these innovations
- increases product differentiation for broadcasters and receiver manufacturers
- provide artistic flexibility
- facilitates international compatibility and enhances the international competitiveness of the US standard
- enhances the opportunity for US based content providers

Within the MPEG-2 transport layer it is quite straightforward to include provision for a Codestream. Consensus on the specification of the minimum target audio decoder should be readily attainable since single chip processors are already available which can run all three (MPEG, DTS and AC-3) of the major multichannel audio decoding algorithms. Due to the

simplicity of traditional audio signal processors it is envisaged that standardization of the Codestream language and syntax will also proceed quickly

American companies today dominate the digital signal processing market, and have also proven to be very innovative in developing audio coding technologies. Adoption of the Codestream approach in the ATSC standard will be highly advantageous to US based companies.